Date: Sun, 10 Apr 94 04:30:24 PDT
From: Ham-Homebrew Mailing List and Newsgroup <ham-homebrew@ucsd.edu>
Errors-To: Ham-Homebrew-Errors@UCSD.Edu
Reply-To: Ham-Homebrew@UCSD.Edu
Precedence: Bulk
Subject: Ham-Homebrew Digest V94 #93
To: Ham-Homebrew

Ham-Homebrew Digest Sun, 10 Apr 94 Volume 94 : Issue 93

Today's Topics:

2 meter transverter for HR 2600??
Cheap skate DDS
Directly plotting etch-resist on PC boards?
Frequency Counter Circuit Mod
How phasing SSB Exciters Work (Was: RF and AF speech processors)
PIN diodes & feedthru caps, WHERE? (2 msgs)
Small (1-5 watt?) AM transmitter.
Small (1-5 watt?) AM transmitter.clo
Summary: adaptive denoising filters for voice

Date: 8 Apr 94 15:03:58 EDT
From: hsdndev!husc-news.harvard.edu!frank!dlleigh@yale.arpa
Subject: 2 meter transverter for HR 2600??
To: ham-homebrew@ucsd.edu

In article <CnuBvM.B2H@comtrol.com> chrise@comtrol.com (Chris Elmquist) writes:
>What I'd like to know-- is if the mentioned HR-2600 can be modified for
>26 to 30 MHz coverage.. or is it likely to be potted with goo that prevents
>you from getting to the appropriate jumpers, diodes or whatever. Very early
>models were modifiable I understand... but recent ones were supposedly
>potted. I've never seen one myself...

There is a company called Chipswitch that sells replacement
microprocessors for the HR-2600 (2510 and Lincoln too) which will allow

the unit to operate from 24.8 MHz to 29.999 MHz and add lots of other features. Great for use with a transverter or if you want to do 12 meters. The chip was about $60 as I recall, and installation was straightforward (though you do have to solder and desolder).

I installed one in mine and it seems to work just fine. Their advertising material claims that some radios may lose PLL lock below 25.5 MHz and they sell a fix for this as well. My 2600 seemed to have no problems all the way down to 24.8 MHz. What I want to figure out now is how to lower the output power so I don't need a huge attenuator between the radio and the transverter.

Their info is:

Chipswitch
4773 Sonoma Hwy. Suite 132
Santa Rosa, CA 95409-4269
(707) 539-0512

Darren Leigh -- kc6euy
dlleigh@frank.harvard.edu

Date: 9 Apr 1994 12:57:13 +0300
From: ihnp4.ucsd.edu!library.ucla.edu!agate!howland.reston.ans.net!pipex!lyra.csx.cam.ac.uk!warwick!uknet!EU.net!news.eunet.fi!news.funet.fi!news.cc.tut.fi!proffa.cc.tut.fi!not-for-mail@@
Subject: Cheap skate DDS
To: ham-homebrew@ucsd.edu

Tom Bruhns (tomb@lsid.hp.com) wrote:

> Alvin Nor Mortensen (mortense@matt.ksu.ksu.edu) wrote:
> : I'm in the process of building a VHF synthesizer. I had originally
> : planned on using DDS to provide the reference to a PLL multiplier.
> : Unfortunately the chip (AD7008 -> Great chip!) doesn't seem to be available
> : with out a significant lead time. I'm considering just rolling my own
> : using the MSB of the phase word as my "DAC". Has anyone tried this?
> : Any problems from the spurs ... ? Any thoughts ...?

> Lots of spurs lurking about out there on this one. Consider a
> clock at 40MHz, used to generate a DDS signal at 10MHz. If it's
> _exactly_ 4:1 division, there won't be any jitter on the output.
> But if you want 10.000 + or - a little, then most of the time

> you divide by 4, but sometimes you divide by 3 or 5, getting to
> the MSB. This means that the spectral purity of the output is
> very poor.

Can anybody estimate the spectral distribution of these spurs ?

The far-out spurs are filtered out by the low-pass filter in the PLL multiplier anyway, so all you have to worry about are the close-in spurs. It depends on the response time requirements of the PLL multiplier how much close-in spurs can be removed.

Paul OH3LWR

Phone   : +358-31-213 3657
X.400   : G=Paul S=Keinanen O=Kotiposti A=ELISA C=FI
Internet: Paul.Keinanen@Telebox.tele.fi
Telex   : 58-100 1825 (ATTN: Keinanen Paul)
Mail   : Hameenpuisto 42 A 26
          FIN-33200 TAMPERE
          FINLAND

Date: 9 Apr 94 19:20:05 GMT
From: sdd.hp.com!hpscit.sc.hp.com!icon!greg@hplabs.hp.com
Subject: Directly plotting etch-resist on PC boards?
To: ham-homebrew@ucsd.edu

Adrian Godwin (agodwin@acorn.co.uk) wrote:
: However, a halfway house (still needs messy etching but getting away from
: the problems with limited resolution of cheaper pens) might be to mill
: away only the etch-resist.

Reminds me of how I used to do PC boards many years ago (when components were big enough to see, and had leads you could count on one hand)... I coated the PC board with candle wax and used a toothpick to scribe around the traces. Warm the board first so the wax flows evenly.

Greg KD6KGW

Date: 9 Apr 94 13:34:05 GMT
From: news-mail-gateway@ucsd.edu
Subject: Frequency Counter Circuit Mod
To: ham-homebrew@ucsd.edu

Hi,
I am repairing my old DSI 600 mhz counter (timebase died).

The unit displays freq readings even when the input signal is
quite small. The readings occuring when input signal is a low
amplitude, are usually slightly higher than the real freq or
just high and wrong and random. This seems normal for this unit.
The Prescaller IC tries to divide by 10 regardless of input
signal amplitude.

I would like to homebrew a circuit mod that inhibits freq display
any time the input signal is below a minimum amplitude that
I would select.

I am looking for any info about how such a circuit is implemented
in modern freq counters, what family of circuits are used to be
usable at 600 mhz.

Suspect some of the 100 series ECL IC's from Motorola, Philips
or Sony may be usable at 600 mhz. Need suggestions where to get
these at low cost in small quantities...just enough for this
project.

Thanks, Dave
w6mik

Date: Fri, 8 Apr 1994 23:05:04 GMT
From: ihnp4.ucsd.edu!swrinde!gatech!newsxfer.itd.umich.edu!news.cic.net!
magnus.acs.ohio-state.edu!csn!col.hp.com!fc.hp.com!wayne@network.ucsd.edu
Subject: How phasing SSB Exciters Work (Was: RF and AF speech processors)
To: ham-homebrew@ucsd.edu

> So long as the additional filtering is done to both channels identically,
> the phase and amplitude matching between the two channels is not afffected.
                          ^^^^^^^^
> AL N1AL

Umm... true. But the overall group delay is affected. There is an offline
discussion going on on this. The essence of it is that the finite zeros out
of the passband by themselves don't affect the phase (certainly true) but
the somewhat different pole positions have an effect of to-be-determined
significance. Stand by.

Wayne

Date: Fri, 8 Apr 1994 18:34:33 GMT
From: ihnp4.ucsd.edu!library.ucla.edu!csulb.edu!csus.edu!netcom.com!
tgm@network.ucsd.edu
Subject: PIN diodes & feedthru caps, WHERE?
To: ham-homebrew@ucsd.edu

Galen Watts (galen@picea.CFNR.ColoState.EDU) wrote:
: I'm in the building mode again, and I'm looking for the above parts via
: mailorder. Mouser, Digi-Key and Allied aren't any help, who is?
: 73, Galen, KF0YJ

I don't know about mail order but you usually can salvage some
feed-through caps from the mechanical tuner section of an old
junk TV. Usually you will find 3 or 4 feed-through caps.

As far as pin diodes go, I've heard rumor of using silicon
rectifier diodes as an ok substitute. Don't use the fast
recovery type. I guess the success of this substitution
is dependent on your frequency of interest. It is more likely to
work at VHF and above.

Thomas KI4N
tgm@netcom.com

Date: Fri, 8 Apr 1994 16:26:06 GMT
From: ihnp4.ucsd.edu!swrinde!sgiblab!wetware!spunky.RedBrick.COM!psinntp!psinntp!
arrl.org!zlau@network.ucsd.edu
Subject: PIN diodes & feedthru caps, WHERE?
To: ham-homebrew@ucsd.edu

Two sources are
Microwave Components of Michigan
PO Box 1697
Taylor MI 48180 313-753-4581 evenings

and Surplus Sales of Nebraska
1502 Jones Str
Omaha NE 68102
402-346-4750.

Both can come up with some pretty exotic RF parts
(not counting European semiconductors--I don't
know anyone selling any sort of selection in the
USA to amateurs)

Galen Watts (galen@picea.CFNR.ColoState.EDU) wrote:
: I'm in the building mode again, and I'm looking for the above parts via
: mailorder. Mouser, Digi-Key and Allied aren't any help, who is?
: 73, Galen, KF0YJ
--
Zack Lau KH6CP/1   2 way QRP WAS
                            8 States on 10 GHz
Internet: zlau@arrl.org   10 grids on 2304 MHz

Date: Sat, 9 Apr 1994 06:17:42 GMT
From: ihnp4.ucsd.edu!usc!howland.reston.ans.net!newsserver.jvnc.net!yale.edu!
cs.yale.edu!nic.smsu.edu!brt581s@network.ucsd.edu
Subject: Small (1-5 watt?) AM transmitter.
To: ham-homebrew@ucsd.edu

CRAIG ALLEN JOHNSTON (cs125410@diff.csc.lsu.edu) wrote:
> Anyway, what I am looking for is plans or a kit for something to transmit
> with. I know how to build a very simple AM transmitter, but it is
> very primitive, and I'm sure my signal would not be very clean or
> steady. So, if anyone knows where I can get a kit or some plans for
> an easily built (by a reasonably mechanical type person with a small
> amt of knowledge of electronics) transmitter, could you point me in the right
> direction?

   Just thought I'd point out that there are a few of us out here who'd
also be very interested in this information. Please, please, _please_
post it here!

   Thanks in advance...

--

                             --Me. (Berry R. Thrailkill)
                                   (brt581s@nic.smsu.edu)

Date: 9 Apr 94 12:02:12 PST
From: ihnp4.ucsd.edu!agate!library.ucla.edu!csulb.edu!nic-nac.CSU.net!clstac!
pkinnes@network.ucsd.edu
Subject: Small (1-5 watt?) AM transmitter.clo
To: ham-homebrew@ucsd.edu

In article <2o51q3$12v@vixen.cso.uiuc.edu>, cs125410@diff.csc.lsu.edu (CRAIG ALLEN
JOHNSTON) writes:

```
-= snip... =-

> steady.  So, if anyone knows where I can get a kit or some plans for
> an easily built (by a reasonably mechanical type person with a small
> amt of knowledge of electronics) transmitter, could you point me in the right
> direction?
>

     Please!  There's quite a bit of interest here at Cal Poly, too, so if
anyone has info, please post here on this group.

Tnx!
--
Grimm

---
Patrick K. Innes
Email:   pkinnes@vmsa.is.csupomona.edu
"Methinks it is like a weasek"
                   -The (infinity - 1)st monkey

Note generic disclaimer:  These views and/or opinions are mine, not those of
any sentient being or group or organization composed thereof.  Thank you.
   ^
  other

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Date: 8 Apr 1994 18:11:24 GMT
From: ihnp4.ucsd.edu!galaxy.ucr.edu!library.ucla.edu!europa.eng.gtefsd.com!
MathWorks.Com!noc.near.net!sunfish.hi.com!brainiac.hi.com!user@network.ucsd.edu
Subject: Summary: adaptive denoising filters for voice
To: ham-homebrew@ucsd.edu

In a recent posting, I asked about the design of adaptive denoising
filters for voice communications. Here is a summary of the post and
the responses.

Many thanks to Randy Cole, Michael E. Deisher, and Steven R. Bryan
for their helpful responses.

Regards,
-Steve

Steve Byan                                   internet: steve@hi.com
Hitachi Computer Products (America), Inc.
1601 Trapelo Road                            phone: (617) 890-0444
Waltham, MA 02154                            FAX: (617) 890-4998
```

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From: steve@hi.com (Steve Byan)
Newsgroups: comp.dsp,rec.radio.amateur.homebrew
Subject: adaptive denoising filters for voice
Followup-To: comp.dsp,rec.radio.amateur.homebrew

Dr. Steven E. Reyer, WA9VNJ, and David L. Hershberger, W9GR published a brief tutorial in the September 1992 issue of "QEX", published by the ARRL. Their adaptive denoising filter uses Widrow's LMS algorithm to adapt an FIR filter, using a delayed version of the input signal as the reference signal. They claim that by proper choice of the delay and adaptation time constants, the FIR filter will adapt to a bandpass response around the portions of the spectrum which have a high "short-term" correlation (corresponding to the speech signal), and thus reduce the uncorrelated noise.

Reyer and Hershberger only reference Widrow et al's 1975 paper "Adaptive Noise Cancelling: Principles and Applications" in the Proc. IEEE Vol 63, No.2. I haven't tracked down this reference yet. (Wish I lived closer to a good university library.)

My questions:
1) Do the other denoising products use an algorithm similar to that described by Reyer and Hershberger? If not, what algorithms do they use?

2) Are there any good tutorial papers on this subject? Any important references other than Widrow et al?

3) It occurs to me that an LPC vocoder, such as used in a CELP vocoder, would serve as an effective denoising filter, since it supposedly models the vocal tract resonances. The excitation for the reconstruction filter would be the input (noisy) signal. Is it feasible to extract the LPC parameters from a noisy signal? Is anyone using this approach?

From: cole@soldev.tti.com (Randy Cole)
Subject: Re: adaptive denoising filters for voice
Date: Mon, 14 Mar 1994 15:16:06

W9GR is on the net and may see your posting.

>My questions:
>1) Do the other denoising products use an algorithm similar to that
>described by Reyer and Hershberger? If not, what algorithms do they use?

There are some that supposedly use a technique you could call "power spectrum gating" or the like. Suppose you compute a long-term average power spectrum. You might convince yourself that this represents a "noise floor". Then you take a segment of the noisy signal, look at the power spectrum, and by some means or another throw away spectral bands that don't exceed the noise floor by some amount.

A grad student friend of mine used this technique 20 years ago and his version
exhibited very noticeable artifacts. Maybe people have thought of better ways
to do it.

>2) Are there any good tutorial papers on this subject? Any important
>references other than Widrow et al?

I dunno. See your (not so local) library :-)

>3) It occurs to me that an LPC vocoder, such as used in a CELP vocoder,
>would serve as an effective denoising filter, since it supposedly models
>the vocal tract resonances. The excitation for the reconstruction filter
>would be the input (noisy) signal. Is it feasible to extract the LPC
>parameters from a noisy signal? Is anyone using this approach?

LPC techniques in general don't handle broadband additive noise all that well. They go to pot pretty quickly for SNR less than 30 dB or so. I'd doubt
that any of those you mention use this approach.

Good luck!

Randy Cole
cole@soldev.tti.com

Date: Wed, 16 Mar 94 09:41:33 MST
From: deisher@enws125.eas.asu.edu (Michael E. Deisher)
Subject: Re: adaptive denoising filters for voice

> 2) Are there any good tutorial papers on this subject? Any important
> references other than Widrow et al?
>
> 3) It occurs to me that an LPC vocoder, such as used in a CELP
> vocoder, would serve as an effective denoising filter, since it

> supposedly models the vocal tract resonances. The excitation for the
> reconstruction filter would be the input (noisy) signal. Is it
> feasible to extract the LPC parameters from a noisy signal? Is
> anyone using this approach?

Yes, with limited success.

Here some tutorial papers on speech enhancement:


```
@incollection{boll92,
  author="S. F. Boll",
  title="Speech Enhancement in the 1980s:  Noise Suppression with
         Pattern Matching",
  booktitle="Advances in Speech Signal Processing",
  editor="S. Furui and M. M. Sondhi",
  publisher="Marcel Dekker",
  year={1992},
  pages={309--325},
  annote="copy on file"}

@article{ephr92c,
  author="Y. Ephraim",
  title="Statistical-Model-Based Speech Enhancement Systems",
  journal="Proceedings of the IEEE",
  volume={80},
  number={10},
  month={October},
  year={1992},
  pages={1526--1555},
  annote="copy on file"}

@article{lim79,
  author="J. S. Lim and A. V. Oppenheim",
  title="Enhancement and Bandwidth Compression of Noisy Speech",
  journal="Proceedings of the IEEE",
  volume={67},
  number={12},
  month={December},
  year={1979},
  pages={1586--1604},
  annote="copy on file"}

@techreport{makh89,
  author="J. Makhoul and T. H. Crystal and D. M. Green and D. Hogan
          and R. J. McAulay and D. B. Pisoni and R. D. Sorkin and
          T. G. Stockham",
  title="Removal of Noise from Noise-Degraded Speech Signals",
  institution="National Research Council",
```

```
  year={1989},
  annote="copy on file"}

@article{osha89,
  author="D. O'Shaughnessy",
  title="Enhancing Speech Degraded by Additive Noise or Interfering
         Speakers",
  journal="IEEE Communications Magazine",
  month={February},
  year={1989},
  pages={46--52},
  annote="copy on file"}
```

A technique similar to the method you attributed to Reyer and
Hershberger may be found in:

J.E. Paul, Automatic digital audio processor (ADAP), Proc. 11th Annual
Asilomar Conf. on Circuits, Systems, and Computers, Pacific Grove, CA,
1978.

Hope this helps.

--Mike

===
 | Mike Deisher   Arizona State University
|
 | deisher@dspsun.eas.asu.edu   Telecommunications Research Center
|
 | voice: (602) 965-0396   Signal Processing Laboratory
|
 | fax: (602) 965-8325   Tempe, AZ 85287-7206
|

===

Date: Thu, 17 Mar 94 10:47:27 -0700
From: sxb@inel.gov (Steve Bryan)
Subject: Re: adaptive denoising filters for voice

Steve,

In article <steve-140394123659@brainiac.hi.com> you write:
>I'm interested in understanding the design of adaptive denoising filters

>for voice communications. There are several manufacturers selling DSP
>denoising filters to the amateur radio marketplace, and apparently some
>commercial units for aviation communications. Some well-known manufacturers
>are JPS Communications, Inc, Timewave Technology Inc, and W9GR.

I own a Timewave DSP-9 with which I am very happy.

>Dr. Steven E. Reyer, WA9VNJ, and David L. Hershberger, W9GR published a
>brief tutorial in the September 1992 issue of "QEX", published by the ARRL.
>Their adaptive denoising filter uses Widrow's LMS algorithm to adapt an FIR
>filter, using a delayed version of the input signal as the reference
>signal. They claim that by proper choice of the delay and adaptation time
>constants, the FIR filter will adapt to a bandpass response around the
>portions of the spectrum which have a high "short-term" correlation
>(corresponding to the speech signal), and thus reduce the uncorrelated
>noise.
>
>Reyer and Hershberger only reference Widrow et al's 1975 paper "Adaptive
>Noise Cancelling: Principles and Applications" in the Proc. IEEE Vol 63,
>No.2. I haven't tracked down this reference yet. (Wish I lived closer to a
>good university library.)

Widrow has published a book entitled "Adaptive Signal Processing". As
of a year ago, it was the only such book that had been published. I own
a copy, but I have only read selected chapters.

>My questions:
>1) Do the other denoising products use an algorithm similar to that
>described by Reyer and Hershberger? If not, what algorithms do they use?

The adaptive filter algorithm is very flexible, and can be used for
all sorts of things. What it does depends on the error signal that you
are providing to the algorithm. I suspect that my DSP-9 uses a
somewhat different approach for noise reduction, still based on adaptive
filtration. The denoiser follows the output of a selectable FIR
bandpass filter, and an optional adaptive notch filter. The adaptive
notch filter is fed a different error signal than the denoiser, and
seems to include more taps in the filter array.

The Timewave documentation is somewhat vague about what the specific
algorithms used are. There is mention of a correlation-based
approach. I think that the peak autocorrelation (not including the
zeroth time lag) of the signal is used to provide the error signal for
adaptation of the denoiser.

>2) Are there any good tutorial papers on this subject? Any important
>references other than Widrow et al?

I think that Widrows book is probably the best on adaptive algorithms and applications. There is a book that is a compendium of articles that is edited by Jae S. Lim called "Speech Enhancement" Prentice-Hall signal processing series, ISBN 0-13-829705-3. It covers other denoising algorithms as applied to speech signals. The table of contents categorizes the articles concerned with enhancement of speech degraded by additive noise under the following topic categories:

Effect of noise on speech communications
Systems based on the perceptial aspects of speech
Systems based on the periodicity of speech
Systems based on an underlying model of speech
Systems that require more than one microphone input
Applications of speech enhancement

There are three other major topics in the book in addition to enhancement of speech degraded by additive noise, which are

Processing of speech prior to its degradation by additive noise
Enhancement of speech degraded by reverberation
Time scale modification of speech

This book is NOT a tutorial, but does contain many basis articles from which many subsequent articles in each of the areas have been published.

>3) It occurs to me that an LPC vocoder, such as used in a CELP vocoder,
>would serve as an effective denoising filter, since it supposedly models
>the vocal tract resonances. The excitation for the reconstruction filter
>would be the input (noisy) signal. Is it feasible to extract the LPC
>parameters from a noisy signal? Is anyone using this approach?

This is covered in Lim's book under "systems based on an underlying model of speech". There are three articles in this section of the book:
"All pole modelling of degraded speech", J.S. Lim and A.V. Oppenheim,
"Linear Predicitive Coding of Speech Signals in a High Ambient Noise Environment", H. Kobatake, J. Inari, and S. Kakuta, and
"Maximum Likelihood Pitch Estimation", J.D. Wise, J.R. Caprio, and T.W. Parks. I kind of doubt that any of the filters used for ham radio use the LPC model approach, since extracting the LPC coefficients is pretty expensive computationally.

Hope that this helps. Let me know if you have any other questions.

73,
-Steve
--

|Steven R. Bryan, Idaho National Engineering Lab. INTERNET: sxb@INEL.GOV
 |
|N7MPY   POB 1625 M.S. 2090   Phone: (208) 525-5484
 |
|_________________Idaho_Falls,_Id._83415___________FAX:___(208)_525-5996_____|

Date: 9 Apr 94 15:39:27 GMT
From: agate!howland.reston.ans.net!cs.utexas.edu!utnut!utgpu!utcsri!
newsflash.concordia.ca!CC.UMontreal.CA!poly-vlsi!nick@ucbvax.berkeley.edu
To: ham-homebrew@ucsd.edu

References <2o3hp5$9b5@acorn.acorn.co.uk>,
<1994Apr8.143042.11376@ke4zv.atl.ga.us>, <2o3vd1$f7h@acorn.acorn.co.uk>r
Subject : Re: Directly plotting etch-resist on PC boards?

In article <2o3vd1$f7h@acorn.acorn.co.uk> agodwin@acorn.co.uk (Adrian Godwin)
writes:

...a pile of stuff deleted...

The way we make test PCBs at work is rather simple and more importantly, QUICK. We draw the pcb artwork using AutoCad (I mostly design microwave stuff) and then send the file to the laser printer at 1:1. I load the laser with a sheet of clear acetate and let the printer do it's thing. Bingo, I have a film positive of my artwork. I then go to the lab and make a contact print on a piece of PCB that has been sensitized with a photo resist coating. Then the PCB goes through the normal proceedure of copper etching, cleaning, etc.

This method has allowed us to make PCB artwork with microstrip lines as small as 7 mil (!) with good results. If you can't load the acetate material into your laser, you can always print to a piece of paper and then photocopy the paper onto a piece of acetate. The only problem here is that most photocopiers do not make perfect 1:1 copies (not close enough for 2.8 GHz microstrip!).

Nick


```
 ****************************************************************************
 *       Nick Ciarallo                                                      *
 *       SR Telecom Inc.           telephone: 514-335-2429  ex: 438         *
 *       Microwave Group           facsimile: 514-334-7783                  *
 *       8150 Trans Canada Hwy     internet : nick@vlsi.polymtl.ca          *
 *       St. Laurent, Quebec       hamradio : ve2hot@ve2fkb.pq.can.na       *
 *       Canada  H4S-1M5                                                    *
 ****************************************************************************
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*   Accept no substitutes, *REAL* ham radio lives on 220 MHz!   *
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End of Ham-Homebrew Digest V94 #93
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